Acoustic Communications 2011 Experiment: Deployment Support and Post Experiment Data Handling and Analysis

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LONG-TERM GOALS

A high performance, versatile, and reliable underwater communications capability is of strategic importance to the U.S. Navy. Operational scenarios involving the use, monitoring, and coordination of multiple undersea assets, both manned and unmanned and mobile and fixed, are significantly enhanced by the ability to communicate quickly, reliably, in a wide range of environmental conditions, and with minimal constraints on the actions of the platforms involved. The long-term goal of the effort in underwater acoustic communications is to integrate research in the areas of physical oceanography, ocean acoustics, signal processing, and communications theory to

1. develop a underwater communications capability that can be deployed on and used by a wide range of platforms with minimal required external infrastructure and which achieves reliable and high performance under a wide range of environmental and operational conditions, and

2. develop an communications performance prediction capability that enables commanders to reliably predict the performance of deployed communications systems as a function of environmental conditions and the location, velocity, and capabilities of the deployed assets.

OBJECTIVES

The objectives of the work under this proposal are to support the Long Term Goals in the context of underwater **acoustic** communications systems. The objectives include

1. Work with Dr. William Hodgkiss, Scripps Institution of Oceanography, to conduct the KAM11 field experiment and gather a significant data set of spatially and temporarily coincident environmental and acoustic data with the acoustic data including receptions of ambient noise, general purpose channel probe signals, and specialized communications signals that are to be evaluated for specific applications. Perform appropriate quality control not the data and make it available to KAM11 participants.

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2. Develop and validate models of the impact of sea surface and upper ocean boundary layer processes on the performance of underwater acoustic communications systems.

3. Develop methods of detecting and synchronizing to known underwater acoustic communications signals at very low signal to noise ratios.

4. Develop, analyze and predict the performance of adaptive decision feedback equalization algorithms that are suitable for use with receive arrays with large numbers of elements and which offer improved performance and reduced computational complexity when compared to standard full rank, hard decision directed adaptive equalization algorithms.

APPROACH

The approach taken to consists of a combination of analysis of field data, the development of new theory for adaptive signal processing algorithms, and the development, testing and analysis of new algorithms. The specifics are described below with paragraph numbers corresponding to the paragraph numbers in the preceding section (OBJECTIVES).

1. The conducting of the KAM11 field experiment was a straight forward exercise in experiment planning and execution with a large group of PIs. The approach was to gather PI requirements signal transmissions and signal and environmental measurements, coordination with the ship and PMRF, working with technicians at WHOI and Scripps to plan deployment and recovery operations, and scheduling of the min-experiments that we incorporated into the overall KAM11 experiment.

   Post-cruise, the approach was to analyze the data in the sequence of transmissions, identify time periods when there were either transmission or reception problems, correct the problems when possible, review the environmental data to determine a representative set of time periods, and then quality check the data in those time periods and make it available to other KAM11 experiment participants.

2. Objective 2 is being pursued in collaboration with Dr. Grant Deane, Scripps Institution of Oceanography and Dr. Andone Lavery, Woods Hole Oceanography. The approach is to develop analytical and numerical models to characterize the channel impulse response from a transmitter to a receiver resulting from surface scattered arrivals. This characterization includes intensity fluctuations due to surface shape and the path length fluctuations due to surface motion. Also included are the effects of near surface bubbles on the intensity and time-variability of the surface scattered arrivals. This characterization is carried out with the analysis of derived expressions, numerical evaluation of expressions, and numerical simulations. The derived expressions and numerical models are validated by comparison between predicted impulse responses and those measured during the KAM11 and SPACE08 field experiments. Environmental measurements taken during those two experiment is used as the inputs to the numerical simulations used to generate the impulse response predictions.

3. The approach taken to achieving low-snr detection of and synchronization to communications signals it to hypothesize that a specific location (time) in a received signal is the starting point for a known detection and synchronization portion of a received signal and train a specially constructed adaptive equalizer to demodulate the signal known signal. If the signal is actually

present and it is a bpsk modulated signal, we would expect that the equalizer would achieve a bit error rate of something less than 0.5 (random guessing for a bpsk signal) indicating the detection of and synchronization to a communication signal. If there is no communications signal present, then the equalizer would be expected to achieve a bit error rate of approximately 0.5. indicating that there has been no detection.

4. Adaptive, multi-channel, phase-coherent equalizers reliably achieve higher performance in terms of achievable data rates in underwater acoustic channels than other forms of signal detection and demodulation algorithms. However, in their customary form they have a large number of parameter that need to be adjusted. This large number of parameters results in both a high computational complexity of the equalizer adaptation algorithm and the need to utilize a long averaging interval in order to achieve stable operation. This long averaging interval decreases the rate of unmodeled acoustic channel fluctuations that can be successfully tracked and compensated for by the equalizer. Finally, the use of such equalizer in turbo-equalization algorithms posses great promise for future performance gains. Yet, in the underwater acoustic environment where the channel can change significantly over the duration of a communications packet. Thus, the equalizer must be adapted in each iteration of the turbo algorithm during which it has imperfect knowledge of the transmitted data symbols which are needed to guide the adaptation algorithm.

   The approach to addressing the "large number of parameters" problem will be to develop beamspace processing methods and adaptive subarray processing methods for transforming the single high dimensional adaptation problem into one or more lower dimensional problems. Supporting results in the field of *Random Matrix Theory* will be developed to assist in the guidance of optimal methods and the prediction and analysis of their performance. The approach to developing an adaptive equalizer capable of operating within the iterative framework of a turbo-equalizer will be to use soft symbol information in the equalizer adaptation process place more weight on symbol decisions which are believed to be reliable and to discount the decisions which are not. The Expectation-Maximize framework will serve as a starting point for this development.

WORK COMPLETED

1. Extensive work has been completed with regard to quality control checking and ensuring accessibility of the KAM11 environmental and acoustic data in WHOI's possession. The thermistor string and CTD data has been downloaded, decoded, and placed in easily readable files available to the other PIs involved in the experiment. The acoustic data has been downloaded. The synchronization of data that was received on the WHOI System 4 (7 km range from the fixed WHOI source) during the first deployment period had been compromised by a power fault in the system. This data has been analyzed and resynchronized and made available to PIs. Received acoustic data has been distributed to all of the WHOI led AComms MURI PIs to support their research efforts. Finally, an overview paper for the KAM11 experiment was co-authored with Dr. William Hodgkiss and presented at the ECUA 2012 meeting in Edinburgh, UK in July, 2012.

2. Completed work on the development and analysis of an environmentally aware beamspace processor and an "effective noise" model for used in the calculate of optimal equalizer weights for a channel estimate based equalizer.

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3. Developed and demonstrated the performance of an equalizer based detector for operation at very low SNRs.

4. Developed and analyzed the performance of an adaptive subarray/subdelay based direct adaptation decision feedback equalizer.

5. Demonstrated a relationship between the angular spread of the signal and noise fields in a communications scenario and the optimal spacing of array elements in an adaptive multi-channel direction adaptation equalizer.

6. Developed new results in random matrix theory to quantify the performance characteristics of adaptive RLS (recursive least squares) based algorithms.

7. Developed and analyzed the performance of a new type of soft information based adaptive DFE.

RESULTS

1. The completed research on beamspace processing techniques for multichannel equalizers identified a class of good beamformer windowing functions that account for the angular spread of the transmit to receive channel. The prolate spheroidal functions are used to create a set of orthogonal beams that span the expected angular spread of the received communications signal for pre-processing the total received signal at a large number of array elements and creating a reduced number of beamspace signals. This technique does not require the real time iterative adaptation of an unconstrained and fully adaptive pre-processing beamformer thus reducing computational complexity and eliminating the occasional observed instability in the iterative adaptation process of the fully adaptive beamformer. The performance loss of the non-adaptive beamformer when compared to the fully adaptive beamformer (when the latter is not exhibiting the instability) is minimal. This creates a more system that has both greater reliability and reduced complexity.

2. The completed work on the effective noise correlation matrix at the output of a channel estimation algorithm (which is the input to a channel estimate based equalizer) modeled this effective noise as the addition of the true ambient noise and the residual signal from the channel estimator. This is the portion of the received signal that is untraceable by the channel estimator which means that it is not accounted for in the channel estimate that is produced and used to calculate the coefficients of the channel estimate based equalizer. The correlation structure of this signal is analyzed and found to have a Toeplitz structure. An adaptive equalizer using the resulting noise correlation matrix shows improved performance when compared to one using an unconstrained matrix.

3. The equalizer based signal detector has shown the ability to reliably detect appropriately constructed signals at in-band SNR of -20 dB and significantly outperform both matched filter and energy detection algorithms. Figure 1 shows the performance comparison between the equalizer based detector and an energy detection algorithm. The equalizer based detector is constructed in a manner to reduce computational complexity and improve performance at low SNRs and the detection signals are constructed to reduce computational complexity. Finally, proper temporal synchronization of the subsequent equalizer used for signal demodulation is a by-product of the detection process.
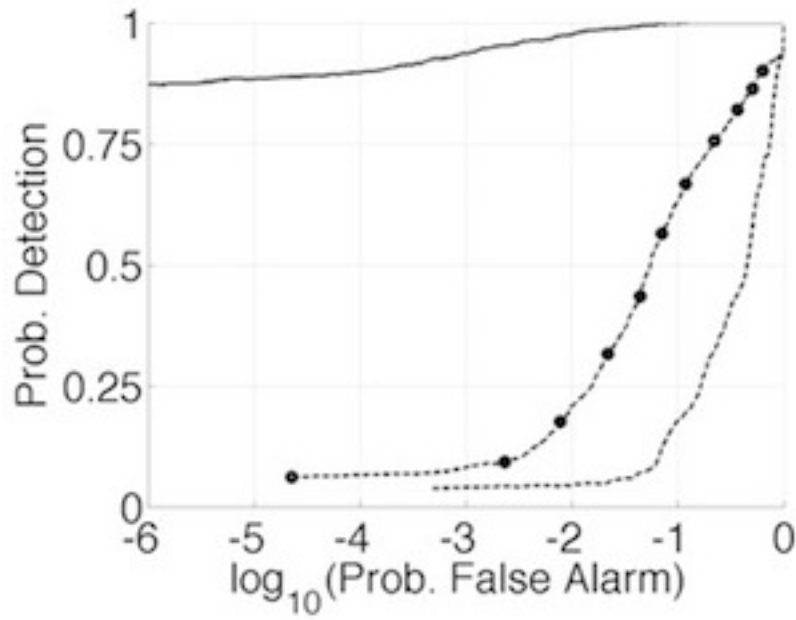
Figure 1: The comparison of the ROCs for the equalizer based detector operating at an in-band signal to noise ratio (SNR) of -20 dB (solid line), an energy detector operating at an SNR of -20 dB (dashed line) and an energy detector operating at an SNR of -10 dB (dashed line with solid circles). This represents results generated from the processing of data from the KAM11 experiments collected at a range of 7 km from the transmitter and over a range of sea surface conditions from calm to rough.

4. Computational complexity and the number of degrees of freedom in an unconstrained multichannel adaptive decision feedback equalizer remain significant problems associated with the widespread use of these algorithms. A subarray/subdelay equalizer has been developed which address both of these challenges. Consider, for example, a multichannel equalizer working on the signals from a 24 channel array. Assume that if the channel is time-invariant and therefore the equalizer filter weights do not need to be adjusted in real-time, then good performance could be achieved with equalizer feedforward filters that have a length of 100 taps (sampled at some multiple of the symbol rate) for each array channel. For purposes of this discussion, we are ignoring the effect of the feedback filter. If we were to use this same equalizer filter configuration in a full array multichannel equalizer in a time-varying environment, there would be 2400 filter tap weights that would need to be adjusted in real-time based upon observations of the channel. The subarray/subdelay (SASD) equalizer would, for example, break the array into 6 subarrays each with 4 channels and would break each 100 tap filter into 4 distinct filters each composed of a contiguous block of 25 taps. Thus, rather than solving one adaptation problem involving 2400 weights the problem is decomposed into 24 independent subproblems each involving 100 weights resulting in significant computational savings and also increasing the rate of channel fluctuations that can be accommodated by the adaptation process. The outputs of the 24 independent subarray/subdelay processors can be adaptively combined in a second stage processor to generate the final equalizer output. The processing of KAM11 data over a range of channel conditions, as expected, significant reductions in computational complexity and improvements in data demodulation performance.

5. The subarray work described previously raises the question of how to partition an array into subarrays in order to achieve the best possible performance. A related problem is to determine the best spacing of array elements given a fixed number of elements. This problem has been investigated empirically and optimal array spacing has been show to be related to the angular spread of the communications signal arrivals at the array as well as the angular spread of the received ambient noise. In surface scattered environments, this can be further related to wind speed and surface conditions and specifically to the presence or absence of a steady state layer of bubbles near the sea surface. The results which enable us to establish this relationship come from Dr. Grant Deane, Scripps Institution of Oceanography. The dependence of optimal spacing upon angular spread comes can be interpreted as a result of spatial aliasing due to the spacing of the array elements beyond half the wavelength of the acoustic signals or in terms of the spatial correlation function of the received signals.

6. New results in the field of random matrix theory which improve our ability to analytically predict the performance of least squares based adaptive filtering algorithms have been developed. The new methods and resulting predictions show a closer match to those achieved in test case simulations than do the predictions based upon previous methods. The new results predict a faster decay in performance as the length of the adaptation algorithm's "averaging interval" is decreased than is predicted by previous method. Interestingly, in some cases, the results also show a moderate improvement in performance followed by a resumption in decaying performance as the "averaging interval" is reduced past a threshold value.

7. A new technique for soft decision directed adaptation of equalizer filter coefficients has been developed. The technique, referred to as Recursive Expected Least Squares, is based upon the Expectation Maximization (E.M.) algorithm and results in improved demodulation performance

when compared to an equalizer using hard symbol decisions in its decision directed adaptation process.

IMPACT/APPLICATIONS

All of the above results are directed towards improving the performance of underwater acoustic communications systems. In particular, these results will allow future systems to operate reliably at lower signal to noise ratios and achieve a higher level of reliability and data transmission rate in challenging environments than is possible with current systems. In addition, the SASD approach will facilitate the implementation of demodulation algorithms on modular, distributed processing hardware architectures that will be readily scalable to working on the data from very large arrays.

RELATED PROJECTS

The work under this grant is closely related to the MURI funded project titled, "Underwater Acoustic Propagation and Communications: A Coupled Research Program", ONR Grant Number N00014-07-10738. Funds from both grants was used to pay for the KAM11 experiment and the post-experiment processing and analysis of the data as well as the development and analysis of channel equalization algorithms.

PUBLICATIONS

M. Pajovic, J. Preisig , "Performance analysis of the least squares based LTI channel identification algorithm using random matrix methods," at *49th Annual Allerton Conference on Communication, Control, and Computing*, Sept. 28-30, 2011, pp. 516-523.

A. Yellepeddi, J. Preisig , "Direct-form adaptive equalization using soft information," at *49th Annual Allerton Conference on Communication, Control, and Computing*, September 28-30, 2011, pp. 461-468.

A. Sen Gupta, J. Preisig, "Adaptive sparse optimization for coherent and quasi-stationary problems using context-based constraints" in *Proc. Int. Conf. on Acoustics, Speech, and Signal Processing*. Kyoto, Japan, March 25 - 30, 2012, pp. 3413 - 3416.

A. Sen Gupta, J. Preisig, A Geometric Mixed Norm Approach to Shallow Water Acoustic Channel Estimation and Tracking, *Elsevier Physical Communication Journal, Special Issue on Compressive Sensing in Communications*, Vol. 5, No. 2, June 2012, pp. 119-128.

Y. Zakharov, A. Morozov, J. Preisig, "Dopper Effect Compensation for Cyclic-Prefix-Free OFDM Signals in Fast-Varying Underwater Acoustic Channel," in *Proc. 11th European Conference on Underwater Acoustics (ECUA 2012)*, July 2 - 6, 2012, pp. 652-659.

W. Hodgkiss, J. Preisig, "Kauai Acomms MURI 2011 (KAM11) Experiment," in *Proc. 11th European Conference on Underwater Acoustics (ECUA 2012)*, July 2 - 6, 2012, pp. 993-1000.

J. Preisig, "Subarray Partitioning for Adaptive Multichannel Equalization," in *Proc. 11th European Conference on Underwater Acoustics (ECUA 2012)*, July 2 - 6, 2012, pp. 1698-1702.

A. G. Deane, J. Preisig, C. Tindle, A. Lavery, M. Stokes, Deterministic forward scatter from surface gravity waves, *JASA*, Submitted, Revised, and Resubmitted.

HONORS/AWARDS/PRIZES

Elected as *Fellow* of the *Acoustical Society of America*.